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Specification and Drawings, as originally filed, with Application for Patent Serial No: **2,214,287**, on August 29, 1997, by **TET HIN YEAP**, for "Method and Apparatus for Encoding a Signal Using Pairs of its Sub-Band Signals for Quadrature Amplitude Modulation".

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S. Gregoire
Agent certificateur/Certifying Officer

September 16, 1998
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ABSTRACT OF THE DISCLOSURE

Improved transmission or storage of signals, such as high speed transmissions in subscriber loops of telecommunication systems, is facilitated by apparatus which includes an encoder for encoding the signal before application to the transmission/storage medium and a decoder which decodes the signal received from the medium. The encoder comprises an analysis filter which analyzes the input signal (Si) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and a device for combining at least one pair of said sub-band signals to provide an encoded signal comprising two adjacent spectral lobes each comprising information from both sub-bands. The decoder comprises a filter for extracting the pair of sub-band signals from a received encoded signal; and a synthesis filter, complementary and substantially inverse to the analysis filter, for processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal. The sub-band signals may be used to modulate in-phase and quadrature components of a common carrier signal to produce the combined signal. One of the spectral components may be removed before transmission/storage of the encoded signal.

METHOD AND APPARATUS FOR ENCODING A SIGNAL USING PAIRS OF ITS SUB-BAND SIGNALS FOR QUADRATURE AMPLITUDE MODULATION

DESCRIPTION

TECHNICAL FIELD:

The invention relates to a method and apparatus for encoding signals, whether digital or analog, for transmission and/or storage. The invention is especially, but not exclusively, applicable to the encoding of digital signals for transmission via communications channels, such as twisted wire pair subscriber loops in telecommunications systems or to storage of signals in or on a storage medium, such as video signal recordings, audio recordings, data storage in computer systems, and so on.

BACKGROUND ART:

Embodiments of the invention are especially applicable to Asynchronous Transfer Mode (ATM) telecommunications systems. Such systems are now available to transmit millions of data bits in a single second and are expected to turn futuristic interactive concepts into exciting realities within the next few years. However, deployment of ATM is hindered by expensive port cost and the cost of running an optical fiber from an ATM switch to the customer-premises using an architecture known as Fiber-to-the-home. Running ATM traffic in part of the subscriber loop over existing copper wires would reduce the cost considerably and render the connection of ATM to customer-premises feasible.

The introduction of ATM signals in the existing twisted-pair subscriber loops leads to a requirement for bit rates which are higher than can be achieved with conventional systems in which there is a tendency, when transmitting at high bit rates, to lose a portion of the signal, typically the higher frequency part, causing the signal quality to suffer significantly. This is particularly acute in two-wire subscriber loops, such as so-called twisted wire pair cables. Using quadrature amplitude modulation (QAM), it is possible to meet the requirements for Asymmetric Digital Subscriber Loops (ADSL), involving rates as high as 1.5 megabits per second for loops up to 3 kilometers long with specified error rates. It is envisaged that ADSL systems will allow rates up to about 8 megabits per second over 1 kilometer loops. Nevertheless, these rates are still considered to be too low, given that standards currently proposed for ATM basic subscriber access involve rates of about 26 megabits per second.

Known QAM systems tend to operate at the higher frequency bands of the channel, which is particularly undesirable for two-wire subscriber loops where attenuation and cross-talk are worse at the higher frequencies. It has been proposed, therefore, to use frequency division modulation (FDM) to divide the transmission system into a set of frequency-indexed sub-channels. The input data is partitioned into temporal blocks, each of which is independently modulated and transmitted in a respective one of the sub-channels. One such system, known as discrete multi-tone transmission (DMT), is disclosed in United States patent specification No. 5,479,447 issued December 1995 and in an article entitled "Performance Evaluation of a Fast Computation Algorithm for the DMT in High-Speed Subscriber Loop", IEEE Journal on Selected Areas in Communications, Vol. 13, No. 9, December 1995 by I. Lee *et al.* Specifically, US 5,479,447 discloses a method and apparatus for adaptive, variable bandwidth, high-speed data transmission of a multi-carrier signal over a digital subscriber loop. The data to be transmitted is divided into multiple data streams which are used to modulate multiple carriers. These modulated carriers are converted to a single high speed signal by means of IFFT (Inverse Fast Fourier Transform) before transmission. At the receiver, Fast Fourier Transform (FFT) is used to split the received signal into modulated carriers which are demodulated to obtain the original multiple data streams.

Such a DMT system is not entirely satisfactory, however, especially for use in two-wire subscriber loops which are very susceptible to noise and other sources of degradation which could result in one or more sub-channels being lost. If only one sub-channel fails, perhaps because of transmission path noise, the total signal is corrupted and either lost or, if error detection is employed, may be retransmitted. It has been proposed to remedy this problem by adaptively eliminating sub-channels, but to do so would involve very complex circuitry.

A further problem with DMT systems is the poor separation between sub-channels. In United States patent specification No. 5,497,398 issued March 1996, M.A. Tzannes and M.C. Tzannes proposed ameliorating the problem of degradation due to sub-channel loss, and obtaining superior burst noise immunity, by replacing the Fast Fourier Transform with a lapped transform, thereby increasing the difference between the main lobe and side lobes of the filter response in each sub-channel. The lapped transform may comprise wavelets, as been disclosed by M.A. Tzannes, M.C. Tzannes and H.L. Resnikoff in an article "The DWMT: A Multicarrier Transceiver for ADSL

using *M*-band Wavelets", ANSI Standard Committee T1E1.4 Contribution 93-067, Mar. 1993 and by S.D. Sandberg, M.A. Tzannes in an article "Overlapped Discrete Multitone Modulation for High Speed Copper Wire Communications", IEEE Journal on Selected Areas in Comm., Vol. 13, No. 9, pp. 1571-1585, Dec. 1995, such systems being referred to as "Discrete Wavelet Multitone (DWMT).

A disadvantage of both DMT and DWMT systems is that they typically use a large number of sub-channels, for example 256 or 512, which leads to complex, costly equipment and equalization and synchronization difficulties. These difficulties are exacerbated if, to take advantage of the better characteristics of the two-wire subscriber loop at lower frequencies, the number of bits transmitted at the lower frequencies is increased and the number of bits transmitted at the higher frequencies reduced correspondingly.

It is known to use sub-band filtering to process digital audio signals prior to recording on a storage medium, such as a compact disc. Thus, US patent specification number 5,214,678 (Rault *et al*) discloses an arrangement for encoding audio signals and the like into a set of sub-band signals using a commutator and a plurality of analysis filters, which could be combined. As shown in their Figures 12 and 13 and described at column 15, lines 5-26, Rault *et al* use recording means which record the sub-band signals as multiple, distinct tracks. This is not entirely satisfactory because each sub-band signal would require its own recording head or, if applied to transmission, its own transmission channel.

United States patent specification number 5,161,210 (Druyvesteyn) discloses a similar analysis technique to that disclosed by Rault *et al* but, in this case, the sub-band signals are combined by means of a synthesis filter before recordal. The input audio signal first is analyzed, and an identification signal is mixed with each of the sub-band signals. The sub-band signals then are recombined. The technique ensures that the identification signal cannot be removed simply by normal filtering. The frequency spectrum of the recombined signal is substantially the same as that of the input signal, so it would still be susceptible to corruption by loss of the higher frequency components. The corresponding decoder also comprises an analysis filter and a synthesis filter. Consequently, the apparatus is very complex and would involve delays which would be detrimental in high speed transmission systems.

It is desirable to combine the sub-band signals in such a way as to reduce the risk of corruption resulting from part of the signal being lost or corrupted during transmission and/or storage.

It should be noted that, although Rault *et al* use the term "analysis filter" in their specification, in this specification the term "analysis filter" will be used henceforth to denote a device which decomposes a signal into a plurality of sub-band signals in such a way that the original signal can be reconstructed using a complementary synthesis filter.

International patent application number .....(Agent's Docket No. AP487PCT).... filed at the Canadian Intellectual Property Office contemporaneously herewith) discloses a method and apparatus for encoding such signals which uses the sub-band signals to modulate carriers which are then combined into a single signal for transmission or storage. According to such international application, various kinds of modulation may be used. However, it has been discovered that quadrature amplitude modulation, when used with sub-band filtered signals, may provide improved operation and reduced complexity/cost.

DISCLOSURE OF INVENTION:

According to one aspect of the present invention, apparatus for encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, comprises:

an encoder comprising

(i) analysis filter means for analyzing the input signal (S_i) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) means for combining at least one pair of said sub-band signals substantially orthogonally to provide a combined signal comprising two orthogonal components each comprising information from both sub-bands, and using said combined signal to provide said encoded signal;

and a decoder comprising

(iii) means for extracting said pair of sub-band signals from a received encoded signal; and

(iv) synthesis filter means complementary and substantially inverse to said analysis filter means for processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

It should be noted that the term "substantially orthogonally" embraces signals which are orthogonal or pseudo-orthogonal.

According to second and third aspects of the invention, there are provided the afore-mentioned encoder *per se* and afore-mentioned decoder *per se*.

According to a fourth aspect of the invention, there is provided a method of encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, comprising the steps of:

at an encoder

(i) using analysis filter means to analyze the input signal (S_i) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) combining at least one pair of said sub-band signals substantially orthogonally to provide a combined signal comprising two orthogonal components each comprising information from both sub-bands, and using said combined signal to provide said encoded signal;

and at decoder, the steps of

(iii) extracting said pair of sub-band signals from a received encoded signal; and

(iv) using synthesis filter means complementary and substantially inverse to said analysis filter means, processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

According to fifth and sixth aspects of the invention, there are provided the afore-mentioned encoding steps *per se* and aforementioned decoding steps *per se*.

BRIEF DESCRIPTION OF THE DRAWINGS:

The foregoing and other objects, features, aspects and advantages of the present invention will become more apparent from the following detailed description, taken in conjunction with the accompanying drawings, of preferred embodiments of the invention, which are described by way of example only.

Figure 1 is a simplified block schematic diagram illustrating a transmission system including an encoder and decoder according to the invention;

Figure 2 is a block schematic diagram of an encoder embodying the present invention;

Figure 3 is a block schematic diagram of a corresponding decoder for decoding signals from the encoder of Figure 1;

Figure 4A illustrates three-stage Discrete Wavelet Transform decomposition using a pyramid algorithm to provide sub-band signals;

Figure 4B illustrates three-stage synthesis of an output signal from the sub-band signals of Figure 4A;

Figure 5 is a block schematic diagram of an encoder using a sub-band analysis filter and quadrature amplitude modulation (QAM) of two sub-bands using components of a single carrier;

Figure 6 is a block schematic diagram of a decoder for decoding signals from the encoder of Figure 5;

Figures 7A, 7B and 7C illustrate the frequency spectrum of an input signal, and two sub-bands before and after quadrature amplitude modulation;

Figure 8 illustrates, as an example, a very simple input signal S_i applied to the encoder of Figure 5;

Figure 9 illustrates the frequency spectrum of the input signal S_i;

Figures 10A, 10B, 10C and 10D illustrate the sub-band signals y_0, y_1, y_2 and y_3, respectively, produced by analysis filtering of the input signal S_i of Figure 8;

Figure 11 illustrates the encoded signal S_0 obtained by modulating sub-band signals y_0 and y_1 using QAM;

Figure 12 illustrates the frequency spectrum of the encoded signal S_o;

Figure 13 illustrates the decoded signal S'_i;

Figure 14 illustrates the frequency spectrum of an encoded signal S''_0 following optional bandpass filtering; and

Figure 15 illustrates the decoded signal S''_i obtained by decoding the bandpass-filtered encoded signal S''_0.

BEST MODES FOR CARRYING OUT THE INVENTION

A transmission system embodying the present invention is illustrated in Figure 1. The system comprises digital input signal source **10**, an encoder **11**, transmission medium **12**, decoder **13** and signal destination **14**. Input signal S_i from signal source **10**

is applied to the encoder **11** which encodes it using sub-band filtering and quadrature amplitude modulation (QAM) and supplies the resulting encoded signal S_o to transmission medium **12**, which is represented by a transmission channel **15**, noise source **16** and summer **17**, the latter combining noise with the signal in the transmission channel **15** before it reaches the decoder **13**. Although a transmission medium is illustrated, it could be an analogous storage medium instead. The output of the decoder **13** is supplied to the signal destination **14**. The usable bandwidth of channel **15** dictates the maximum allowable rate of a signal that could be transmitted over the channel.

A first embodiment of the encoder **11** is illustrated in more detail in Figure 2. The input signal S_i is applied via an input port **20** to analysis filter bank **21** which decomposes it into sub-bands to generate/extract a lowpass sub-band signal y_0, bandpass sub-band signals y_1 - y_{N-2} and a highpass sub-band signal y_{N-1}. The sub-band signals y_1 - y_{N-1} are supplied to a multi-carrier modulator **22** which uses selected pairs of the sub-band signals to modulate a respective carrier of a selected frequency, as will be explained later. The lowpass sub-band signal y_0 and first bandpass sub-band signal y_1 contain more low frequency components than the other sub-band signals so that pair is used to modulate a low frequency carrier f_0. The bandpass sub-band signals y_2 - y_{N-2} and highpass sub-band signal y_{N-1} are used to modulate higher frequency carrier signals f_1 - $f_{N/2}$, respectively, of which the frequencies increase from f_1 to $f_{N/2}$. The resulting modulated carrier signals $y'_{0,1}$ - $y'_{(N-2),(N-1)}$ are combined by summer **23** to form the encoded output signal S_o which is transmitted via output port 24 to transmission medium **12** for transmission to decoder **13** (Figure 1).

For reasons which will be explained later, the output of the summer **23** may be supplied to transmission medium **12** by way of a filter **25**, as shown in broken lines. In this particular example, filter **25** is a bandpass filter.

A suitable decoder **13**, for decoding the encoded output signal S_o, will now be described with reference to Figure 3. After passing through the transmission medium **12**, the transmitted signal S_o may be attenuated and contain noise. Hence, as received by way of port **30** of the decoder **13**, it is identified as received signal S'_o (the prime signifying that it is not identical to encoded signal S_o) and supplied to a filter array **31**. Each of the filters in the array **31** corresponds to one of the frequencies f_o - $f_{N/2}$ of the multi-carrier modulator **22** (Figure 2) and recovers the corresponding modulated carrier signals. The recovered modulated carrier signals $y''_{0,1}$ - $y''_{(N-2),(N-1)}$ separated by the array

31 are demodulated by a multi-carrier demodulator 32 to recover lowpass, bandpass and highpass sub-band signals y^*_0 - y^*_{N-1} corresponding to sub-band signals y_0 - y_{N-1} in the encoder 11. These recovered sub-band signals are supplied to synthesis filter bank 33 which, operating in a complementary and inverse manner to analysis filter bank 21, produces an output signal S'_i, which should closely resemble the input signal S_i in Figure 2, and supplies it to signal destination 14 via output port 34. Usually, the signal S'_i will be equalized using an adaptive equalizer (not shown) to compensate for distortion and noise introduced by the channel 12.

It should be noted that some of the sub-band signal pairs in Figure 2 may not need to be transmitted, if they contain little transmission power as compared with other sub-band signals. When these sub-band signals are not transmitted, the synthesis filter bank 33 shown in Figure 3 will insert "zero" level signals in place of the missing sub-band signals. The reconstructed signal S'_1 would then be only a close approximation to the original input signal S_i. Generally, the more sub-bands used, the better the approximation.

Preferably, analysis filter 21 (Figure 2) is a multiresolution filter bank which implements a Discrete Wavelet Transform (DWT) such as is disclosed in Canadian patent application number 2,184,541 and International patent application No. ...(Agent's ref. AP487PCT)... filed at the Canadian Receiving Office on August 29, 1997, to which the reader is directed for reference.

In order to facilitate a better understanding of the embodiments which use DWT, a brief introduction to discrete wavelet transforms (DWT) will first be given. DWT represents an arbitrary square integrable function as the superposition of a family of basis functions called *wavelets*. A family of wavelet basis functions can be generated by translating and dilating the *mother wavelet* corresponding to the family. The DWT coefficients can be obtained by taking the inner product between the input signal and the wavelet functions. Since the basis functions are translated and dilated versions of each other, a simpler algorithm, known as *Mallat's tree algorithm* or *pyramid algorithm*, has been proposed by S. G. Mallat in "A theory of multiresolution signal decomposition: the wavelet representation", *IEEE Trans. on Pattern Recognition and Machine Intelligence*, Vol. 11, No. 7, July 1989. In this algorithm, the DWT coefficients of one stage can be calculated from the DWT coefficients of the previous stage, which is expressed as follows:

$$W_L(n,j) = \sum_m W_L(m,j-1)\,h(m-2n) \tag{1a}$$

$$W_H(n,j) = \sum_m W_L(m,j-1)\,g(m-2n) \tag{1b}$$

where $W(p,q)$ is the p-th wavelet coefficient at the q-th stage, and $h(n)$ and $g(n)$ are the dilation coefficients corresponding to the scaling and wavelet functions, respectively.

For computing the DWT coefficients of the discrete-time data, it is assumed that the input data represents the DWT coefficients of a high resolution stage. Equations 1a and 1b can then be used for obtaining DWT coefficients of subsequent stages. In practice, this decomposition is performed for only a few stages. It should be noted that the dilation coefficients $h(n)$ represent a lowpass filter, whereas the coefficients **g(n)** represent a highpass filter. Hence, DWT extracts information from the signal at different scales. The first stage of wavelet decomposition extracts the details of the signal (high frequency components) while the second and all subsequent stages of wavelet decomposition extract progressively coarser information (lower frequency components). It should be noted that compactly supported wavelets can be generated by a perfect-reconstruction two-channel filter banks with a so-called octave-band tree-structured architecture. Orthogonal and biorthogonal filter banks can be used to generate wavelets in these system. A three stage octave-band tree structure for Discrete Wavelet Transformation will now be described with reference to Figures 4A and 4B, in which the same components in the different stages have the same reference number but with the suffix letter of the stage.

Referring to Figure 4A, the three decomposition stages A, B and C have different sampling rates. Each of the three stages A, B and C comprises a highpass filter **40** in series with a downsampler **41**, and a lowpass filter **42** in series with a downsampler **43**. The cut-off frequency of each lowpass filter **42** is substantially the same as the cut-off frequency of the associated highpass filter **40**. In each stage, the cut-off frequency is equal to one quarter of the sampling rate for that stage.

The N samples of input signal S_i are supplied in common to the inputs of highpass filter **40A** and lowpass filter **42A**. The corresponding N high frequency samples from highpass filter **40A** are downsampled by a factor of 2 by downsampler **41A** and the

resulting N/2 samples supplied to the output as the highpass wavelet y_3. The N low frequency samples from lowpass filter **42A** are downsampled by a factor of 2 by downsampler **43A** and the resulting N/2 samples supplied to stage B where the same procedure is repeated. In stage B, the N/2 higher frequency samples from highpass filter **40B** are downsampled by downsampler **41B** and the resulting N/4 samples supplied to the output as bandpass wavelet y_2. The other N/2 samples from lowpass filter **42B** are downsampled by downsampler **43B** and the resulting N/4 samples are supplied to the third stage C, in which highpass filter **40C** and downsampler **41C** process them in like manner to provide at the output N/8 samples as bandpass wavelet y_1. The other N/4 samples from lowpass filter **42C** are downsampled by downsampler 43C to give N/8 samples and supplies them to the output as low-pass wavelet y_0.

It should be noted that, if the input signal segment comprises, for example, 1024 samples or data points, wavelets y_0 and y_1 comprise only 128 samples, wavelet y_2 comprises 256 samples and wavelet y_3 comprises 512 samples.

Instead of the octave-band structure of Figure 4A, a set of one lowpass, two bandpass filters and one highpass filter could be used, in parallel, with different downsampling rates.

Referring now to Figure 4B, in order to reconstruct the original input signal, the DWT wavelet signals are upsampled and passed through another set of lowpass and highpass filters, the operation being expressed as:

$$W_L(n,j) = \sum_k W_L(k,j+1)\,h'(n-2k) + \sum_l W_H(l,j+1)\,g'(n-2l) \qquad (2)$$

where $h'(n)$ and $g'(n)$ are, respectively, the lowpass and highpass synthesis filters corresponding to the mother wavelet. It is observed from equation 2 that j-th level DWT wavelet signals can be obtained from $(j+1)$-th level DWT coefficients.

Compactly supported wavelets are generally used in various applications. Table I lists a few orthonormal wavelet filter coefficients ($h(n)$) that are popular in various applications as disclosed by I. Daubechies, in "Orthonormal bases of compactly supported wavelets", Comm. Pure Appl. Math, Vol. 41, pp. 906-966, 1988. These wavelets have the property of having the maximum number of vanishing moments for a given order, and are known as "Daubechies wavelets".

Coefficients	Wavelets	
	Daub-6	Daub-8
h(0)	0.332671	0.230378
h(1)	0.806892	0.714847
h(2)	0.459878	0.630881
h(3)	-0.135011	-0.027984
h(4)	-0.085441	-0.187035
h(5)	0.035226	0.030841
h(6)		0.032883
h(7)		-0.010597

Table I

An embodiment of the invention in which the higher sub-bands are not transmitted, and which uses discrete wavelet transforms for encoding a digital signal, will now be described with reference to Figure 5. In the encoder **11'** of Figure 5, the input signal S_i is supplied via input port **20** to an octave-band filter bank **51** which applies a Discrete Wavelet Transform to the signal S_i to generate lowpass sub-band wavelet signal y_0, two bandpass sub-band wavelet signals, y_1 and y_2, and the highpass sub-band wavelet signal y_3. In this implementation, only sub-band wavelet signals y_0 and y_1 are processed. Bandpass wavelet sub-band signal y_2 and highpass sub-band wavelet signal y_3 are discarded. Interpolator means **52**, interpolates each of the pair of sub-band wavelet signals y_0 and y_1, respectively, by the same factor M, where M is an integer, typically 8 to 24. Thus within interpolator **52**, the sub-band wavelet signals y_0 and y_1 are upsampled by upsamplers 53_0 and 53_1, respectively, which insert zero value samples at intervals between actual samples. The upsampled signals then are filtered by two Raise-Cosine filters 54_0 and 54_1, respectively, which insert at each upsampled "zero" point a sample calculated from actual values of previous samples. The Raise-Cosine filters are preferred so as to minimize intersymbol interference. The two interpolated sub-band wavelet signals y''_0 and y''_1 are supplied to quadrature amplitude modulator **55** which uses them to modulate in-phase and quadrature components f_I and f_Q of a carrier signal f_0, provided by oscillator **56**, the quadrature component being derived by means of a phase

shifter **57**. The modulator **55** comprises multipliers **58_0** and **58_1** which multiply the carrier signal in-phase and quadrature components f_I, and f_Q by the two interpolated sub-band wavelet signals y_0, and y_1, respectively. The resulting two modulated carrier signals y'_0 and y'_1 are added together by a summer **59** to form the encoded signal S_o for transmission by way of port **24** (and bandpass filter **25** if provided) to transmission medium **12**.

At the corresponding decoder 13' shown in Figure 6, the signal S'_o received at port **30** is supplied to a QAM demodulator **61** which comprises multipliers **62_0** and **62_1**, which multiply the signal S'_o by in-phase and quadrature components f_I and f_Q of a carrier signal f_0 from an oscillator **63**, the quadrature signal f_Q being derived by way of phase shifter **64**. The resulting signals are passed through lowpass filters **65_0** and **65_1**, respectively, which extract the upsampled versions y^{u*}_0 and y^{u*}_1 which then are decimated by decimators **66_0** and **66_1**, respectively, of decimator **67**. The resulting recovered sub-band signals y^*_0 and y^*_1 are each supplied directly to a corresponding one of two inputs of a synthesis filter bank **68** which applies to them an Inverse Discrete Wavelet Transform (IDWT) as per Figure 4B to recover the signal S'_i which corresponds to the input signal S_i. The highpass sub-band wavelet signals y_2 and y_3, which were not transmitted, are replaced by a "zero" signal at the corresponding "higher" frequency inputs **69_2** and **69_3** of the synthesis filter bank **68**. The resulting output signal S'_i from the synthesis filter bank **68** is the decoder output signal supplied via output port **34**, and is a close approximation to the input signal S_i at the input to the encoder **11'** of Figure 5.

If the higher sub-band signals are used, the DQAM and decimator would be duplicated as appropriate and a suitable synthesis filter used.

Figures 7A to 15 illustrate simplified signals at various points in the system during operation and, in some, the frequency spectrum. Figure 7A shows the frequency spectrum of a much-simplified input signal S_i occupying a bandwidth **BW** centered at frequency f_c. As shown in Figure 7B, after analysis filtering and interpolation, the input signal S_i has been partitioned into two interpolated sub-band signals, y^u_0 and y^u_1. It should be noted that, for complex input signals, the sub-band signals y_0 and y_1 prior to interpolation have a very wide spectrum. After upsampling and filtering by the interpolator **52** (Figure 5), sub-band signals y_0 and y_1 each have a spectrum that is narrower than the frequency spectrum of the original signal S_i Theoretically, their

bandwidth **BW'** is substantially equal to one half of the bandwidth **BW** of the original signal.

As shown in Figure 7C, following modulation by the QAM means **55**, the output signal from the QAM means **55** has a spectrum which has two lobes, one each side of the carrier frequency $\mathbf{f_0}$ used by the QAM means **55**. The center frequency of the lower frequency lobe is equal to $\mathbf{f_0} - \Delta$, and the center frequency of the upper frequency lobe is equal to $\mathbf{f_0}$ plus Δ, where Δ preferably is equal to about one quarter of the bandwidth **BW** of the original input signal S_i. However, Δ may vary depending upon the complexity of the input signal and the design of the analysis filter **51**. The bandwidth **BW'** is determined in dependence upon the sampling rate of the digital input signal S_i.

In the aforementioned Canadian patent application number 2,184,541 and corresponding PCT application, the sub-bands were modulated onto separate carriers so their frequency spectrum lobes were separated by a guard band and each lobe contained information from its own sub-band only. By contrast, in the present invention, there is no need for a guard band between the lobes in the output signal S_o. (Figure 7C). It should be noted that each lobe contains information from both of the sub-band signals y_0 and y_1. Thus, as illustrated in Figures 7A and 7B, the information **A** contained in the input signal S_i is split into lower-frequency component **L(A)** in sub-band signal y_o and higher-frequency component **H(A)** in sub-band signal y_1. As shown in Figure 7C, after quadrature amplitude modulation, each lobe of encoded signal S_o contains some of components **L(A)** and **H(A)**. Consequently, if one lobe is corrupted, perhaps because of noise or attenuation of higher frequencies, it may still be possible to reconstruct the original signal S_i. Hence, the transmission is more robust.

It should be appreciated that, if signal compression is desired, perhaps because bandwidth is limited, one of the lobes need not be transmitted. If the lower lobe were to be discarded, a high pass filter could be used to filter the output from encoder **11** in Figure 5. Conversely, if the higher lobe were to be discarded, a low pass filter could be used instead. In the embodiments shown in Figures 2 and 5, a bandpass filter **25** is shown (in broken lines) for removing the higher lobe. Use of a bandpass filter rather than a low-pass filter allows the portion of the spectrum below and above the lower lobe to be used for other purposes.

It should be noted that this is not the same as single sideband transmission where, although each sideband contains the same information, it is derived from a single source via a single modulated carrier.

Figure 8 illustrates, in the time domain, a very simple input signal S_i comprising two sinusoidal signals, of 400 Hz and 1200 Hz, respectively. Figure 9 illustrates the corresponding frequency spectrum of this two-tone input signal S_i.

Figures 10A, 10B, 10C and 10D illustrate the corresponding four sub-band signals y_0, y_1, y_2 and y_3, respectively, obtained by analysis filtering the input signal. It should be noted that bandpass sub-band signal y_2 has little energy compared with signals y_0 and y_1 and the energy content of highpass sub-band signal y_3 is negligible. Hence sub-band signals y_2 and y_3 are not used in encoding the encoded signal S_o which is illustrated in Figure 11. As shown in Figure 12, the frequency spectrum of the encoded signal S_o comprises two lobes, with respective peaks at 1600 Hz and 2400 Hz, i.e. at an offset Δ of 400 Hz either side of a center frequency of 2000 Hz. Some bandpass filtering was applied to remove harmonics.

Figure 13 illustrates the corresponding decoded signal S'_i and shows that the two tones of the original input signal S_i have been recovered, but without the portion corresponding to omitted sub-band signal y_2. It will be appreciated that, with suitable adaptive equalisation, the original digital signal can be recovered despite a portion of the signal (sub-band signal y_2) not being transmitted.

As mentioned previously, a further reduction in bandwidth can be achieved by transmitting only one lobe of the encoded signal, predicated upon the fact that each lobe contains information from both sub-bands. Thus, Figure 14 illustrates the frequency spectrum of the encoded signal S''_o when filtered by bandpass filter **25** to remove the higher-frequency lobe. Figure 15 illustrates the corresponding decoded signal S'_i and shows that, despite the fact that one lobe was not transmitted, the two tones have been recovered by the decoder.

Embodiments of the invention which allow higher frequency components and lower frequency components to be intermixed and compressed into a narrower bandwidth than the original signal are especially useful for use with two-wire subscriber loops of telecommunications systems since such loops tend to attenuate higher frequencies disproportionately.

It should be noted that the sub-band signal bandwidths could be greater than one half of the original signal bandwidth **BW**, though still less. This would allow a less expensive analysis filter to be used.

It should be appreciated that, where the analysis filter **21** implements **DWT**, the synthesis filter **33** will implement inverse **DWT**.

It should be appreciated that the quadrature amplitude modulation means **55** could comprise a Carrierless Amplitude/Phase (CAP) modulation means which would comprise an in-phase filter means and a quadrature filter means each of which integrates the interpolation of the corresponding sub-band signal with the multiplier function, in essence combining interpolator **52** and QAM **55** (Figure 5).

While similar implementations using more than two pairs of sub-bands and carriers are possible, and might be desirable in some circumstances, for most applications, and especially communication of digital signals via twisted wire subscriber loops, they would be considered complex without significant improvement in performance.

It is envisaged that, instead of bandpass filter **25**, other means could be used to eliminate one of the lobes of the encoded signal before transmission/storage. For example, filter **25** could be replaced by a Fast Fourier Transform device, a phase shifting and cancellation circuit, or other suitable means.

Although a multiresolution filter, specifically one implementing **DWT** is preferred, other forms of analysis filter could be used instead.

If more sub-band signal pairs were to be used, an interpolator **52** and QAM **55** would be provided for each additional pair, which would also be interpolated at such a rate that all of the modulated carriers had the same bit rate. For a large number of sub-bands, it might then be preferable to use a uniform analysis filter bank rather than a multiresolution analysis filter bank.

INDUSTRIAL APPLICABILITY

It should be appreciated that the signal source **10** and the encoder **11** could be parts of a transmitter having other signal processing circuitry. Likewise, the decoder **13** and signal destination **14** could be parts of a corresponding receiver.

It should be noted that the present invention is not limited to transmission systems but could be used for other purposes to maintain signal integrity despite noise and

attenuation. For example, it might be used in recording of the signal on a compact disc or other storage medium. The storage medium can therefore be equated with the transmission medium **12** in Figure 1. It should be appreciated that the encoders and decoders described herein would probably be implemented by a suitably programmed digital signal processor or as a custom integrated circuit.

Although embodiments of the invention have been described and illustrated in detail, it is to be clearly understood that the same is by way of illustration and example only and not to be taken by way of the limitation, the spirit and scope of the present invention being limited only by the appended claims.

CLAIMS:

1. Apparatus for encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, comprising:

an encoder comprising

(i) analysis filter means for analyzing the input signal (Si) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) means for combining a pair of said sub-band signals to provide a said encoded signal having two spectral lobes each comprising information from both sub-bands of said pair;

and a decoder comprising

(iii) means for extracting said pair of sub-band signals from a received encoded signal; and

(iv) synthesis filter means complementary and substantially inverse to said analysis filter means for processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

2. Apparatus as claimed in claim 1, wherein the combining means comprises modulation means for using said pair of sub-band signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal;

and the decoder comprises demodulation means for extracting the sub-band signals from the received encoded signal.

3. Apparatus as claimed in claim 2, wherein the modulation means comprises interpolation means for interpolating the sub-band signals, and quadrature amplitude modulation means for using each of the interpolated sub-band signals to modulate a respective one of an in-phase carrier signal and a quadrature carrier signal, the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and the demodulation means comprises

means for demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

4. Apparatus as claimed in claim 1, wherein the analysis filter generates a plurality of pairs of sub-band signals and the modulation means modulates a selection of said pairs, the synthesis filter compensating for the unused sub-band signals by substituting zero level signals.

5. Apparatus as claimed in any one of claims 1 to 4, further comprising means for removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

6. An encoder for encoding an input signal for transmission or storage comprising:

(i) analysis filter means for analyzing the input signal (Si) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) means for combining at least one pair of said sub-band signals to provide a said encoded signal comprising two spectral lobes each comprising information from both sub-bands.

7. An encoder as claimed in claim 6, wherein the combining means comprises modulation means for using said pair of sub-band signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal.

8. An encoder as claimed in claim 7, wherein the modulation means comprises interpolation means for interpolating the sub-band signals, and quadrature amplitude modulation means for using each of the interpolated sub-band signals to modulate a respective one of an in-phase carrier signal and a quadrature carrier signal, the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other,

9. An encoder as claimed in claim 6, wherein the analysis filter generates a plurality of pairs of sub-band signals and the modulation means modulates a selection of said pairs.

10. An encoder as claimed in any one of claims 6 to 9, further comprising means for removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

11. A decoder for decoding an encoded signal encoded by the encoder of claim 6, comprising:

(iii) means for extracting said pair of sub-band signals from a received encoded signal; and

(iv) synthesis filter means complementary and substantially inverse to said analysis filter means for processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

12. A decoder as claimed in claim 11, for decoding an encoded signal encoded by the encoder of claim 7 and further comprising demodulation means for extracting the sub-band signals from the received encoded signal.

13. A decoder as claimed in claim 12, for decoding an encoded signal encoded by the encoder of claim 8, and wherein the demodulation means comprises means for demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

14. A decoder as claimed in claim 13, for decoding an encoded signal encoded by the encoder of claim 9, and wherein the synthesis filter is arranged to compensate for the unused sub-band signals by substituting zero level signals.

15. A method of encoding an input signal for transmission or storage and decoding such encoded signal to reconstruct the input signal, comprising the steps of:
at an encoder

(i) using analysis filter means to analyze the input signal (Si) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) combining at least one pair of said sub-band signals to provide a said encoded signal comprising two spectral lobes each comprising information from both sub-bands;

and at decoder, the steps of

(iii) extracting said pair of sub-band signals from a received encoded signal; and

(iv) using synthesis filter means complementary and substantially inverse to said analysis filter means, processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

16. A method as claimed in claim 15, wherein the combining step comprises using said pair of sub-band signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal;
and the demodulation at the decoder comprises the step of extracting the sub-band signals from the received encoded signal.

17. A method as claimed in claim 6, wherein the modulation comprises the step of interpolating the sub-band signals, and using quadrature amplitude modulation means for using each of the interpolated sub-band signals to modulate a respective one of an in-phase carrier signal and a quadrature carrier signal, the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other, and the demodulation step at the decoder comprises the step of demodulating the received encoded signal using in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

18. A method as claimed in claim 15, wherein a plurality of pairs of sub-band signals are generated but only a selection of said pairs modulated, and the processing by the synthesis filter means compensates for the unused sub-band signals by substituting zero level signals.

19. A method as claimed in any one of claims 15 to 18, further comprising the step of removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

20. A method of encoding an input signal for transmission or storage comprising the steps of:

(i) using analysis filter means to analyze the input signal (Si) into a plurality of sub-band signals (y), each sub-band centered at a respective one of a corresponding plurality of frequencies (f); and

(ii) combining at least one pair of said sub-band signals to provide a said encoded signal comprising two spectral lobes each comprising information from both sub-bands.

21. An encoding method as claimed in claim 20, wherein the combining step comprises the step of using said pair of sub-band signals each to provide a respective one of a first modulated signal and a second modulated signal, the first modulated signal and the second modulated signal having the same frequency but phase displaced by 90 degrees one relative to the other, and combining the modulated signals to provide said encoded signal.

22. An encoding method as claimed in claim 21, wherein the modulation comprises the step of interpolating the sub-band signals, and using each of the interpolated sub-band signals for quadrature amplitude modulation means of a respective one of an in-phase carrier signal and a quadrature carrier signal, the in-phase carrier signal and the quadrature carrier signal having the same frequency but phase-displaced by 90 degrees one relative to the other.

23. An encoding method as claimed in claim 20, wherein a plurality of pairs of sub-band signals are generated using the analysis filter means but only a selection of said pairs modulated.

24. An encoding method as claimed in any one of claims 20 to 23, further comprising the step of removing one of said spectral lobes from the encoded signal and providing the remaining one of said spectral lobes as said encoded signal.

25. A method of decoding an encoded signal encoded by the encoder of claim 20, comprising the steps of:

(iii) extracting said pair of sub-band signals from a received encoded signal; and

(iv) using synthesis filter means complementary and substantially inverse to said analysis filter means, processing the extracted pair of sub-band signals to produce a decoded signal corresponding to the input signal.

26. A decoding method as claimed in claim 25, for decoding an encoded signal encoded by the encoding method of claim 21 and further comprising the step of demodulating the received signal to extract the sub-band signals.

27. A decoding method as claimed in claim 26, for decoding an encoded signal encoded by the encoding method of claim 22, and wherein the demodulating of the received encoded signal uses in-phase and quadrature carrier signals having the same frequency as those used to encode the encoded signal.

28. A decoding method as claimed in claim 25, for decoding an encoded signal encoded by the encoder of claim 23, and wherein the processing using the synthesis filter is arranged to compensate for the unused sub-band signals by substituting zero level signals.

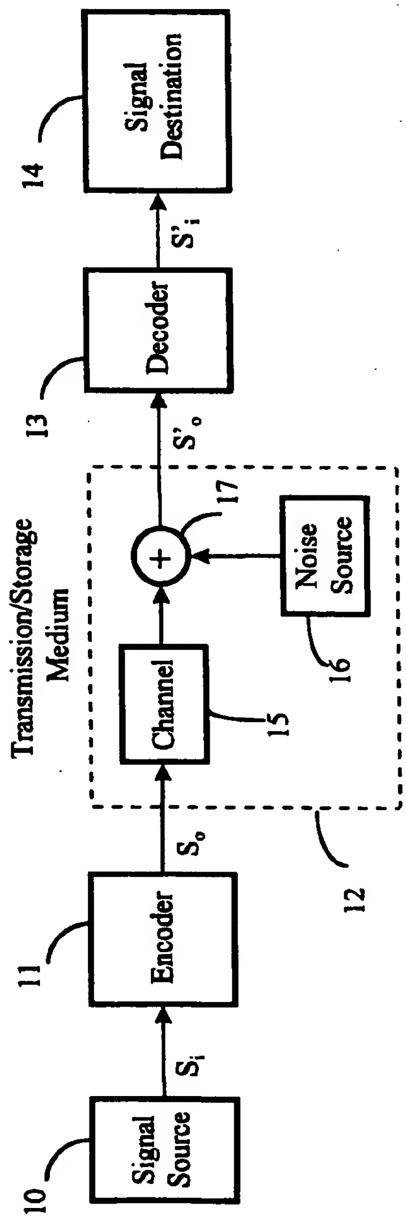

Signal Source
10
Si
Encoder
11
So
Transmission/Storage Medium
Channel
15
+
17
Noise Source
16
12
S'o
Decoder
13
S'i
Signal Destination
14

FIG. 1

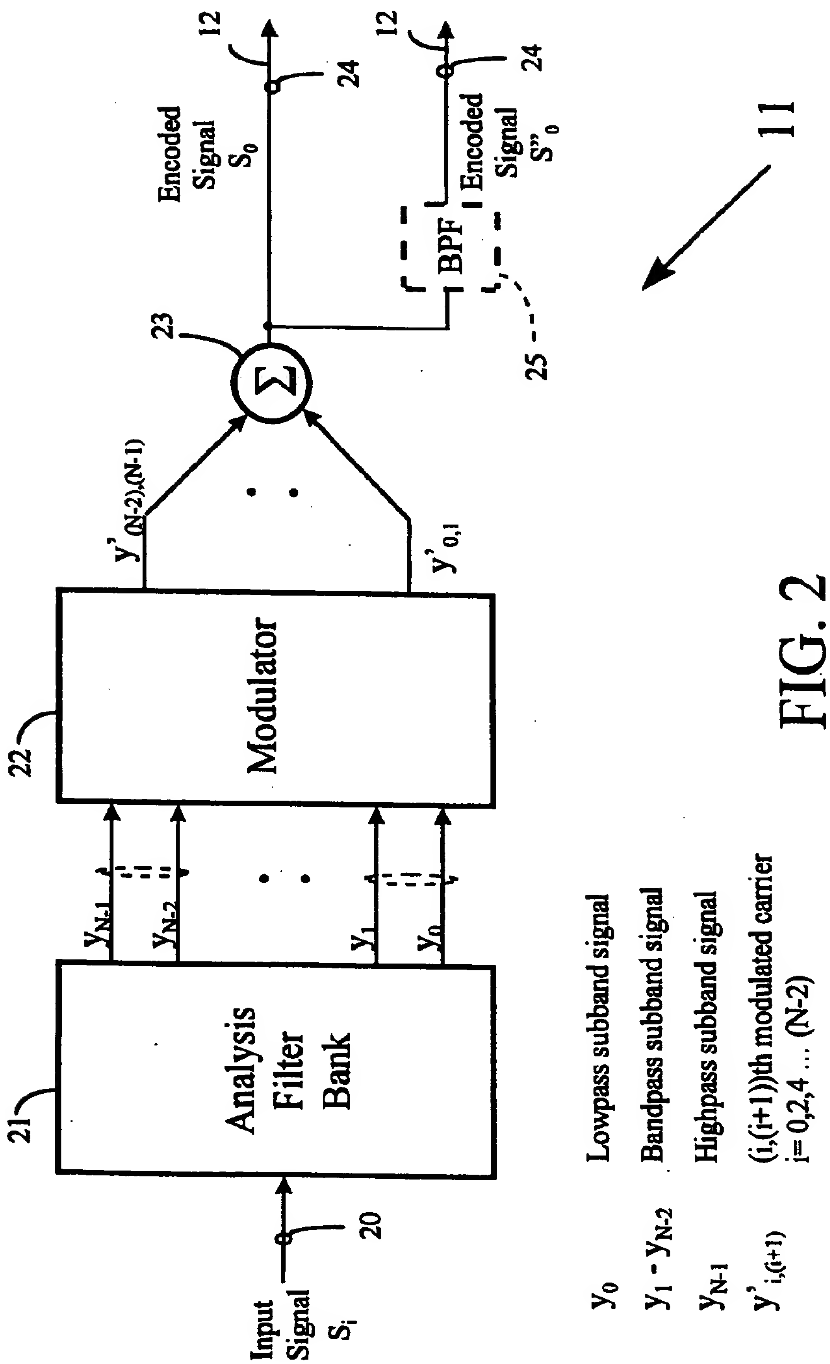
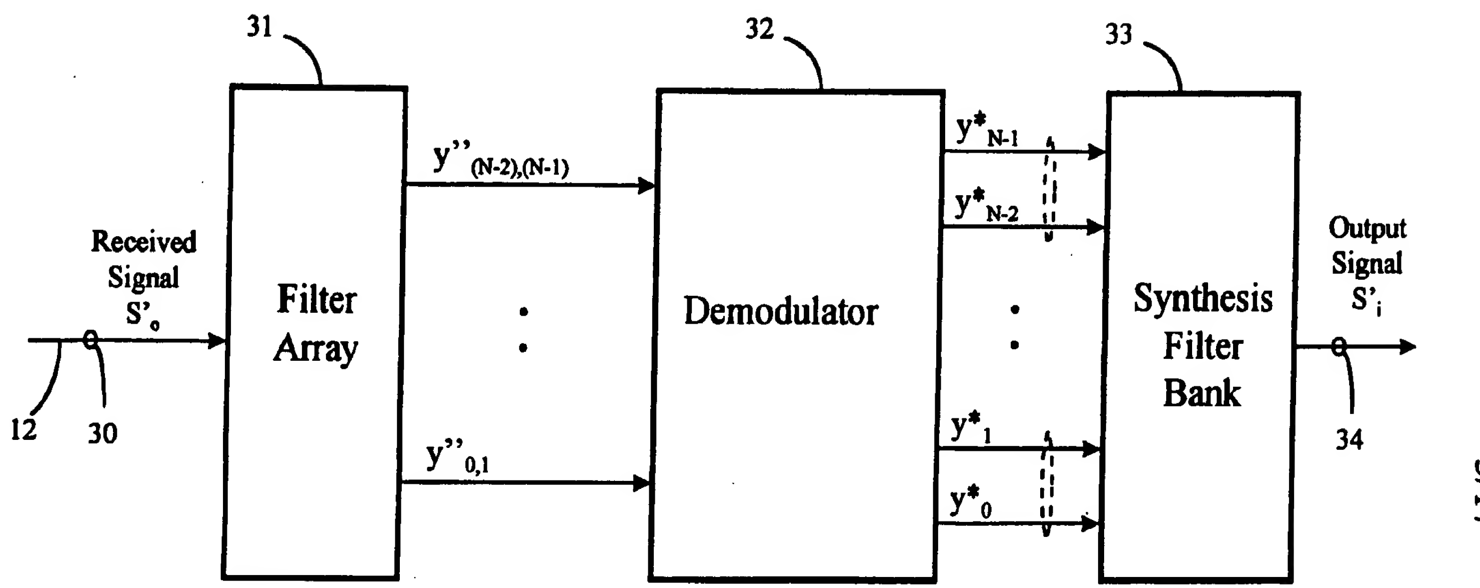

Input Signal S_i
20
21
Analysis Filter Bank
y_{N-1}
y_{N-2}
y_1
y_0
22
Modulator
$y'_{(N-2),(N-1)}$
$y'_{0,1}$
23
Σ
Encoded Signal S_0
24
12
BPF
25
Encoded Signal S''_0
24
12
11
y_0 Lowpass subband signal
$y_1 - y_{N-2}$ Bandpass subband signal
y_{N-1} Highpass subband signal
$y'_{i,(i+1)}$ (i,(i+1))th modulated carrier
i= 0,2,4 ... (N-2)

FIG. 2



y^*_0	Recovered lowpass subband signal
$y^*_1 - y^*_{N-2}$	Recovered bandpass subband signal
y^*_{N-1}	Recovered highpass subband signal
$y''_{i,i+1}$	Recovered (i,(i+1))th modulated carrier i = 0,2,4 ... N-2

FIG. 3

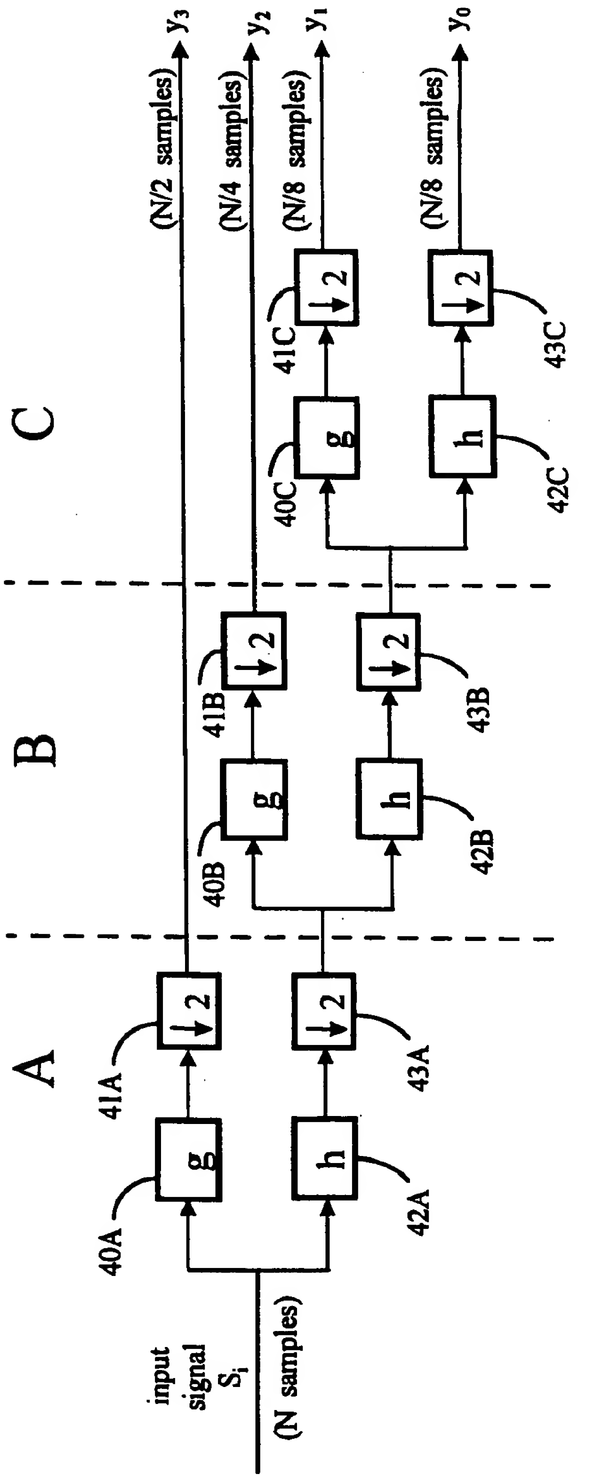

A
B
C
input signal S_i
(N samples)
g
h
↓2
40A
41A
42A
43A
40B
41B
42B
43B
40C
41C
42C
43C
(N/2 samples)
(N/4 samples)
(N/8 samples)
(N/8 samples)
y_3
y_2
y_1
y_0

FIG. 4A

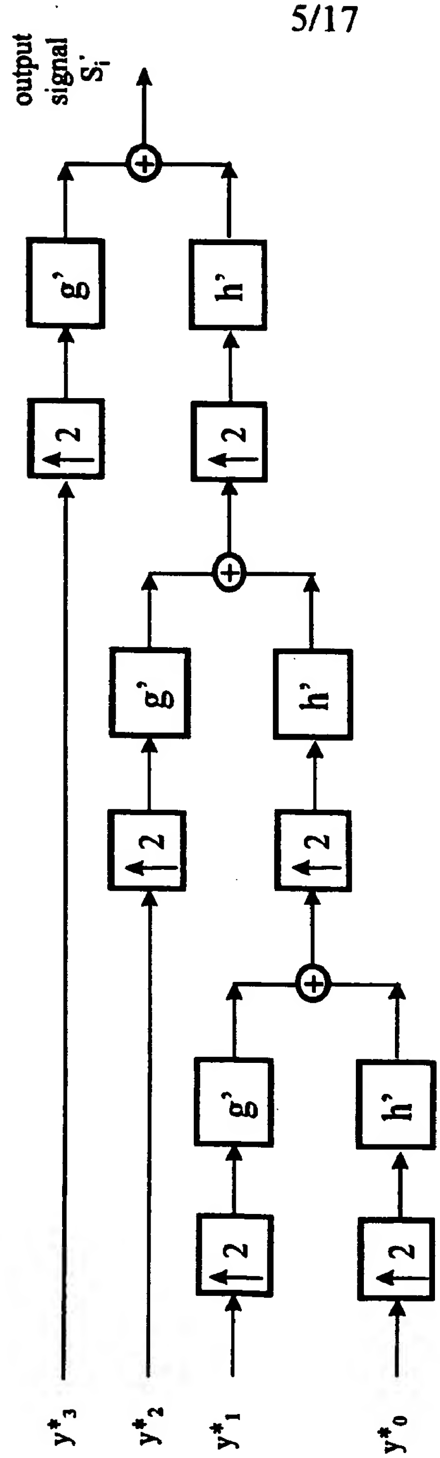


FIG. 4B

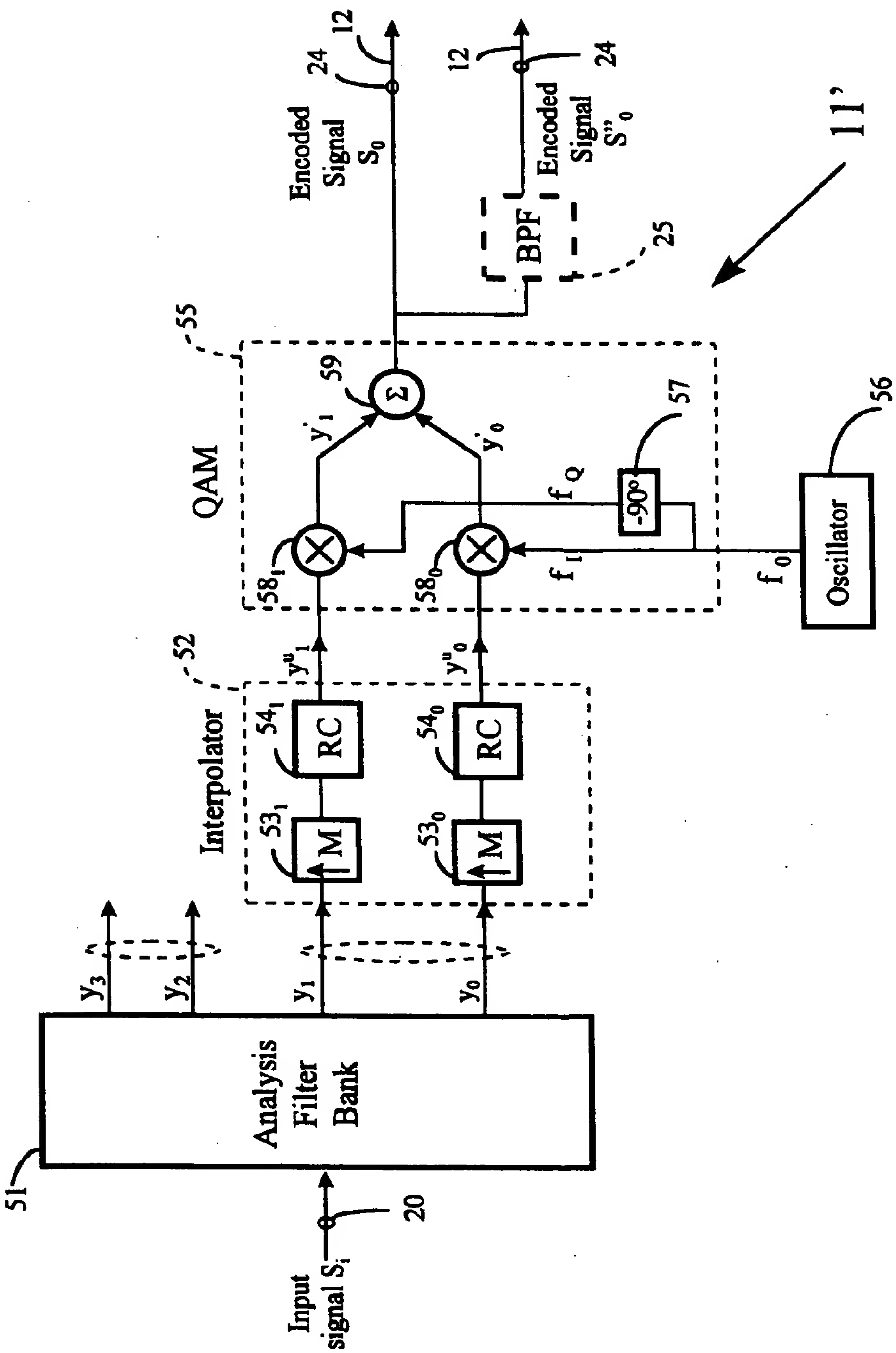

Input signal S_i
20
51
Analysis Filter Bank
y_0
y_1
y_2
y_3
Interpolator
52
53_1
M
54_1
RC
53_0
M
54_0
RC
y^u_1
y^u_0
QAM
55
58_1
58_0
f_I
f_Q
-90°
57
Σ
59
y'_1
y'_0
Oscillator
56
f_0
BPF
25
Encoded Signal S_0
Encoded Signal S''_0
24
12
24
12
11'

FIG. 5

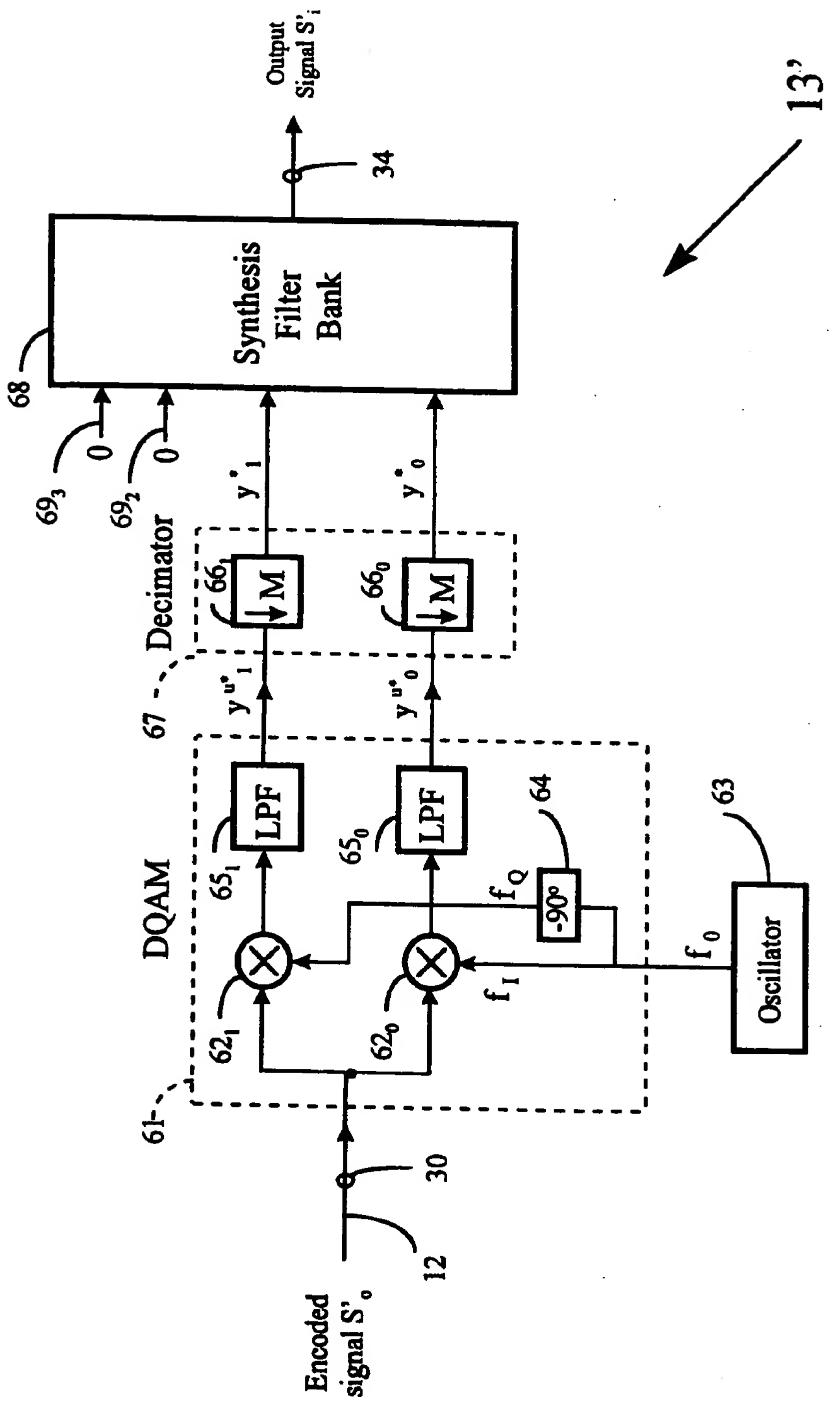
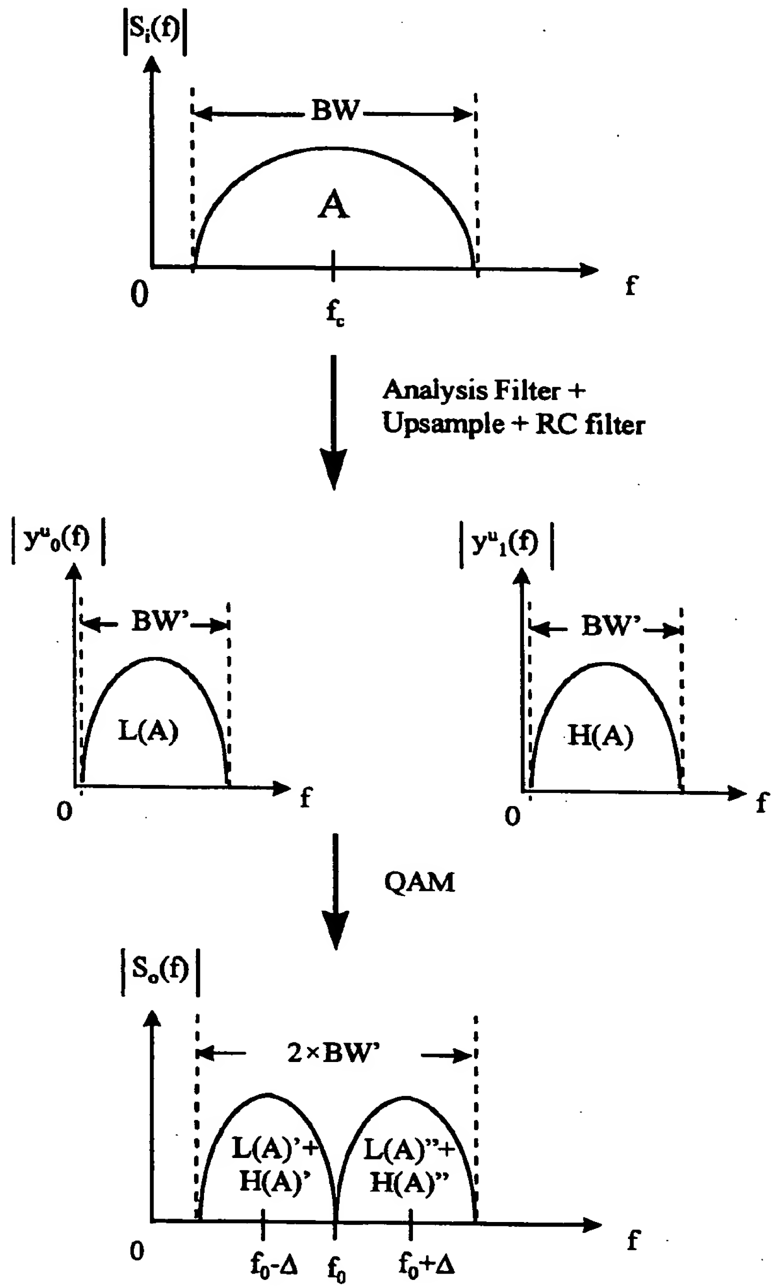

Output Signal S'i
34
Synthesis Filter Bank
68
13'
693
692
0
0
y*1
y*0
Decimator
661
660
M
67
yu*1
yu*0
DQAM
LPF
LPF
651
650
64
-90°
fQ
fI
f0
63
Oscillator
621
620
61
30
12
Encoded signal S'o

FIG. 6

FIG. 7A

FIG. 7B

FIG. 7C



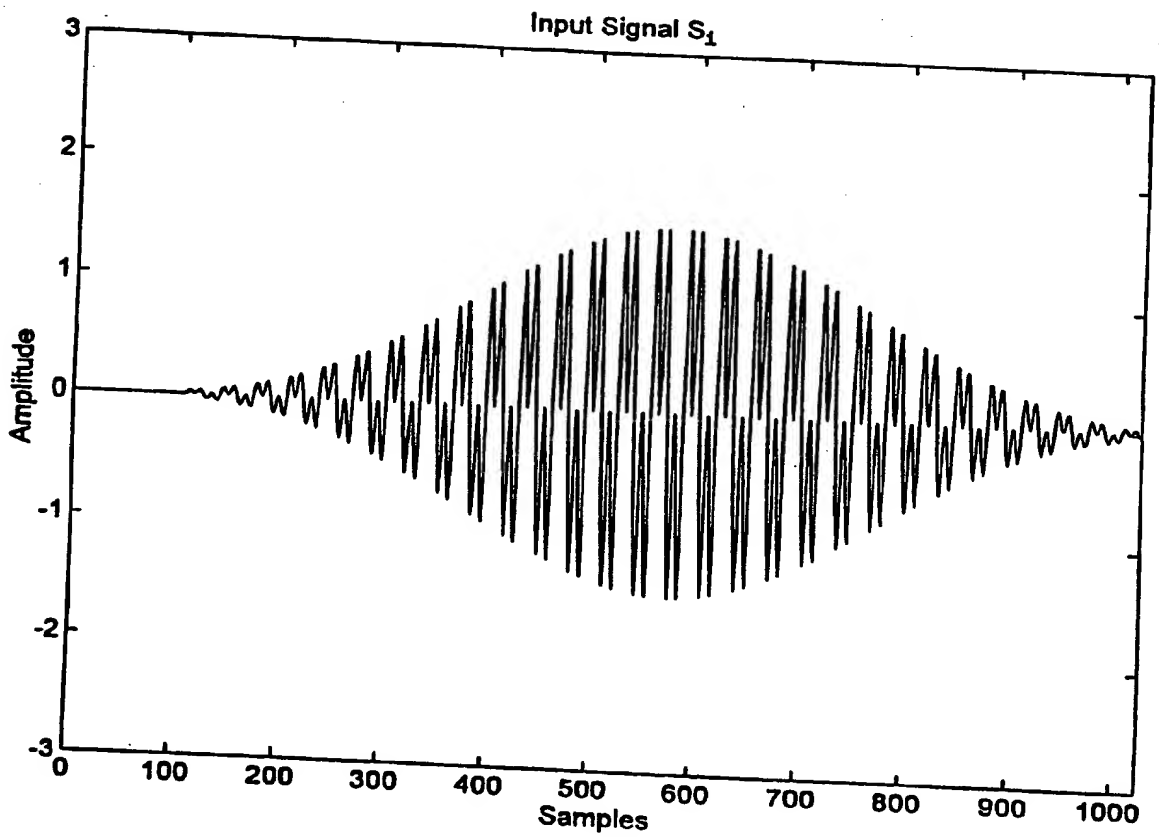
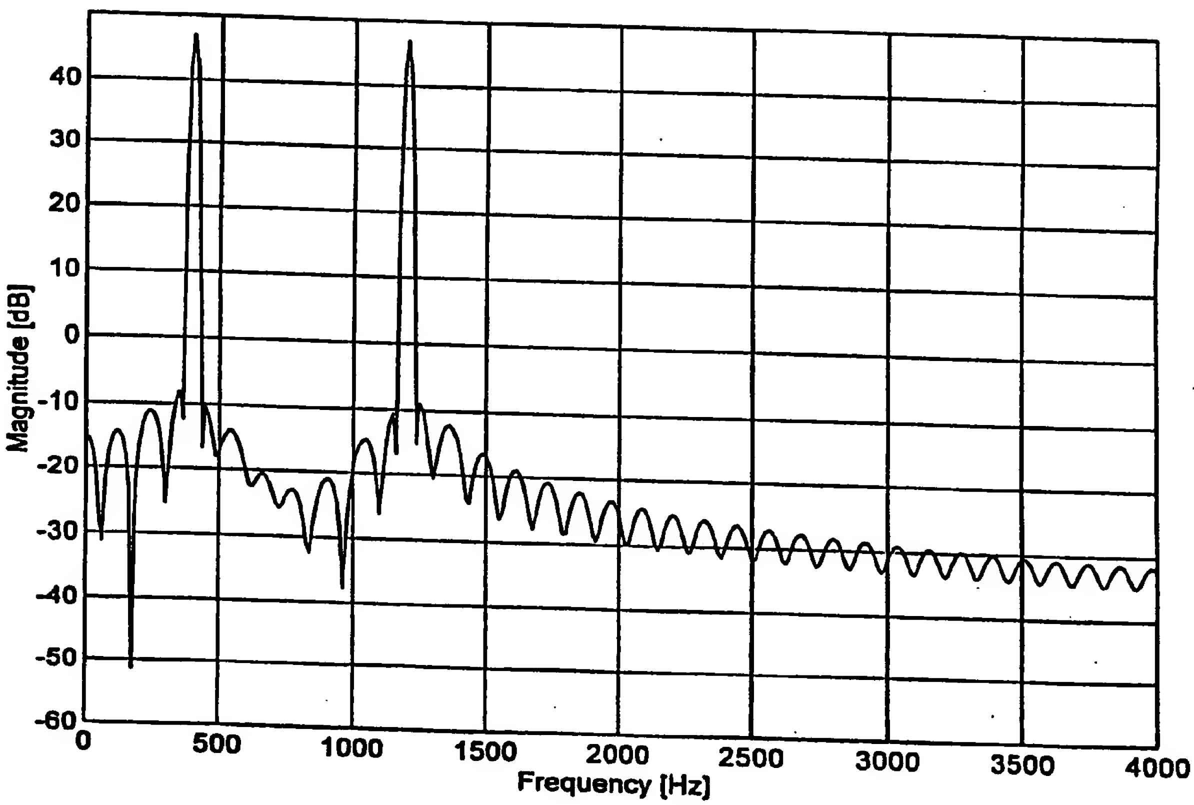

Input Signal S_1
Amplitude
Samples
3
2
1
0
-1
-2
-3
0
100
200
300
400
500
600
700
800
900
1000

FIG. 8



40
30
20
10
0
-10
-20
-30
-40
-50
-60
Magnitude [dB]
0
500
1000
1500
2000
2500
3000
3500
4000
Frequency [Hz]

FIG. 9

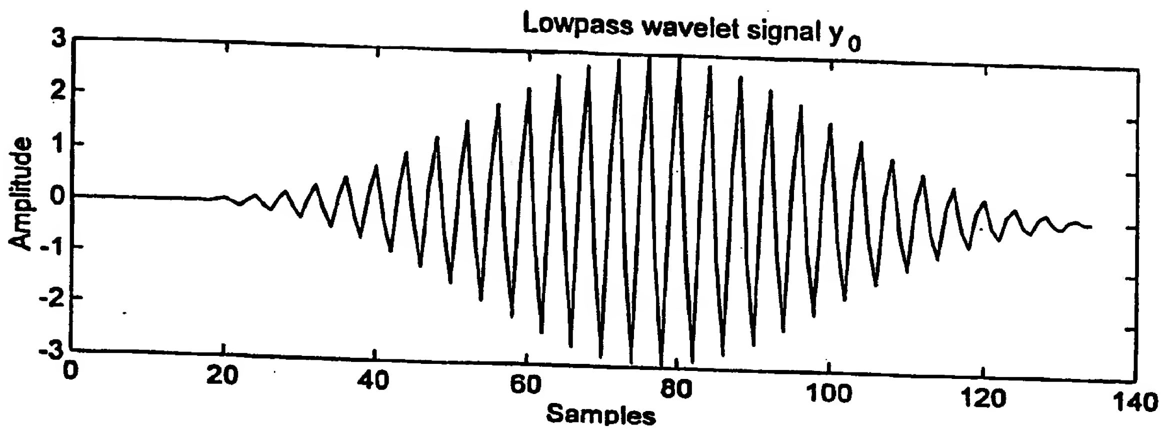


FIG. 10A

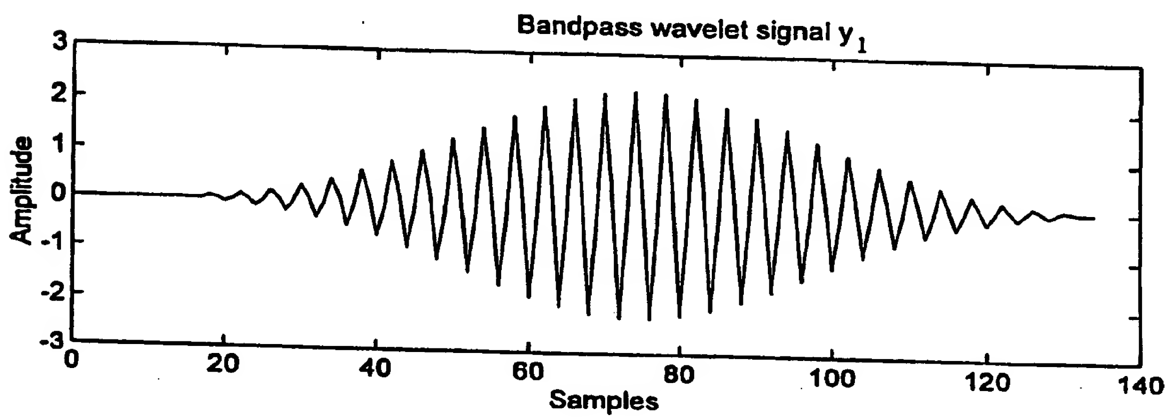


FIG. 10B

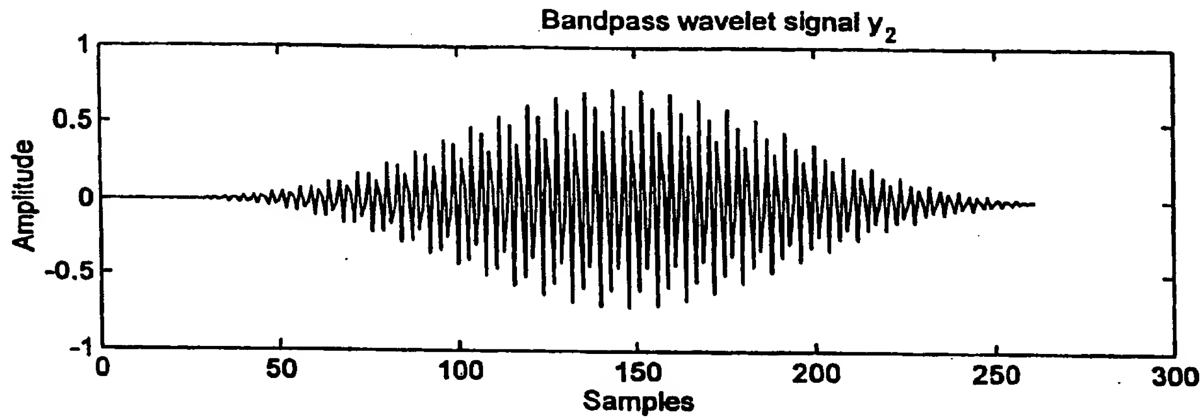


FIG. 10C

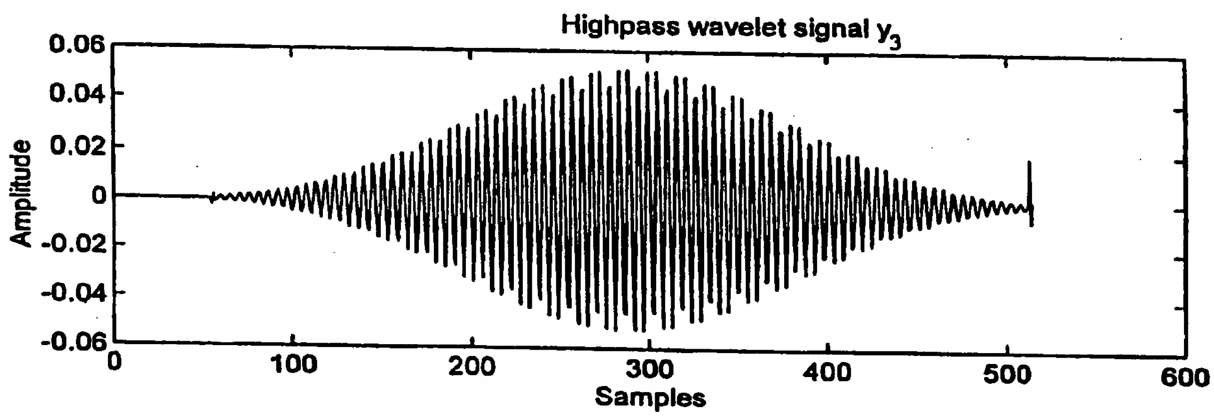


FIG. 10D

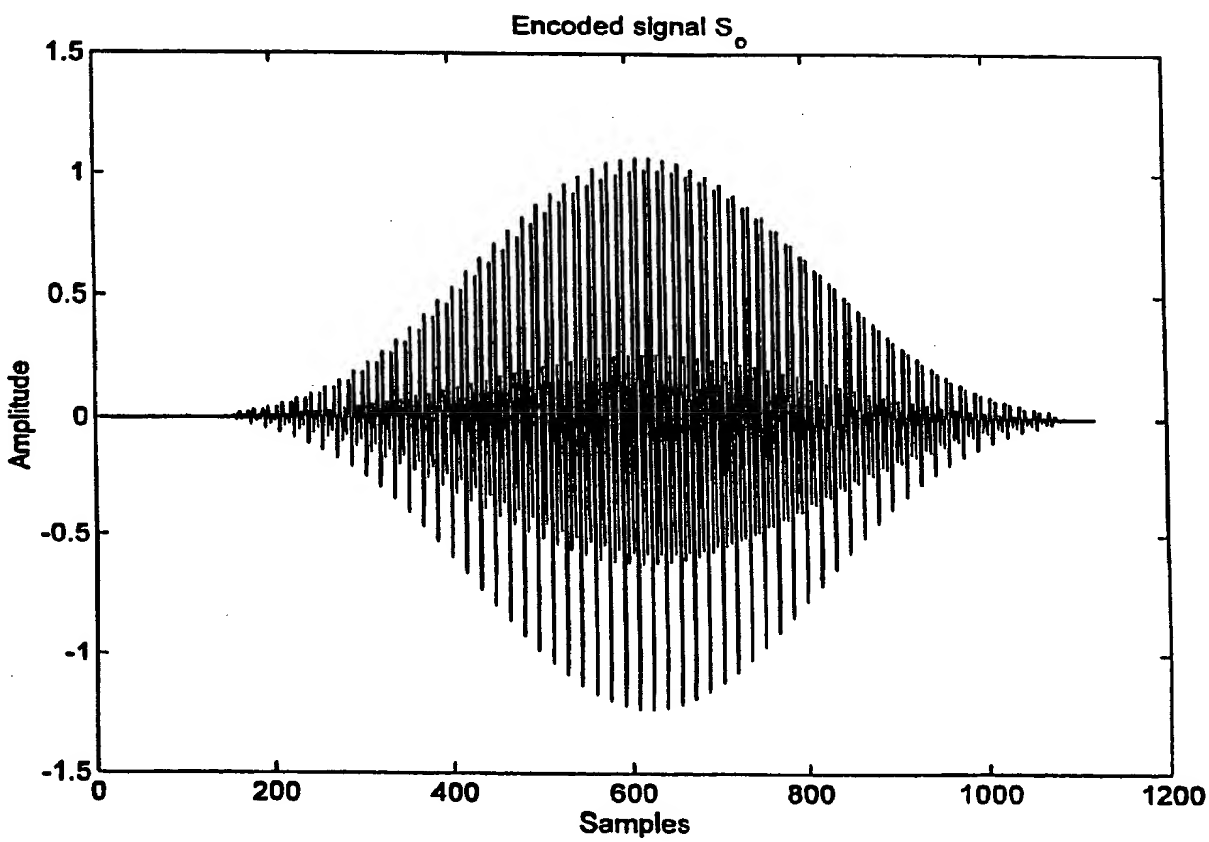

Encoded signal S_o
1.5
1
0.5
0
-0.5
-1
-1.5
Amplitude
0
200
400
600
800
1000
1200
Samples

FIG. 11

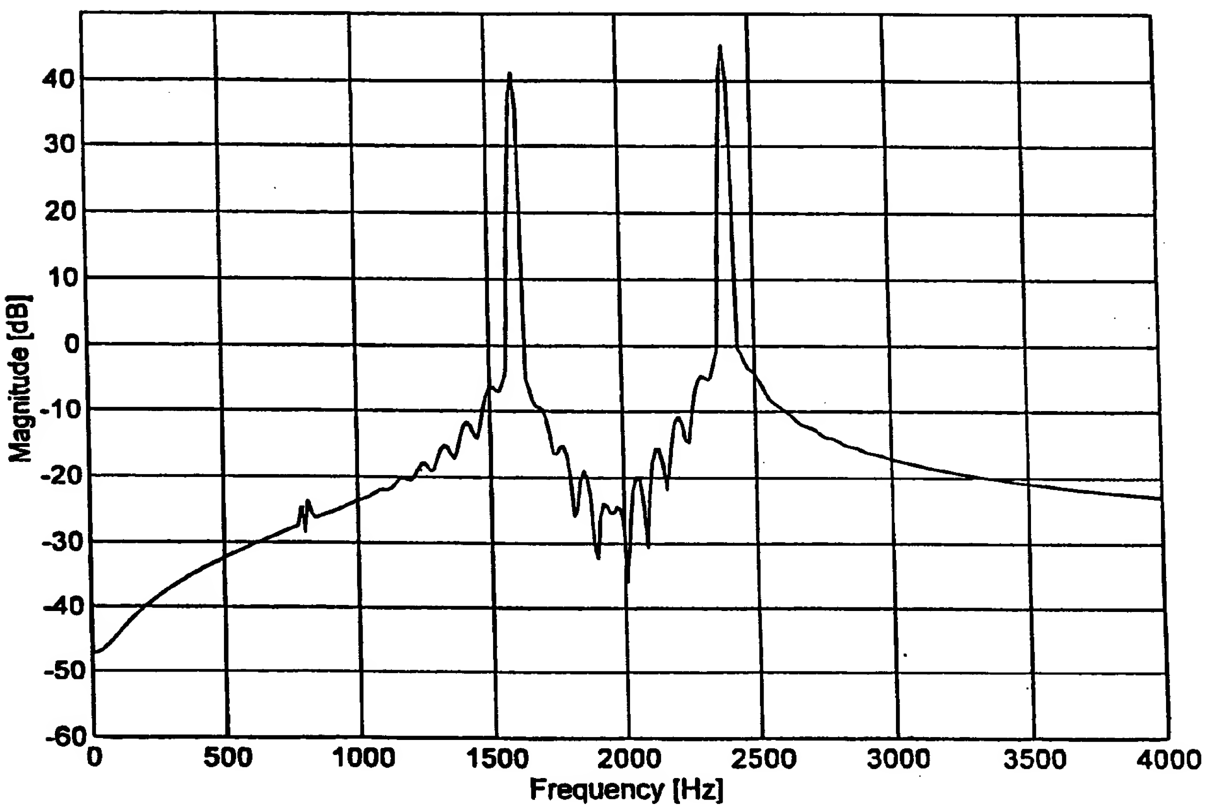

40
30
20
10
0
-10
-20
-30
-40
-50
-60
Magnitude [dB]
0
500
1000
1500
2000
2500
3000
3500
4000
Frequency [Hz]

FIG. 12

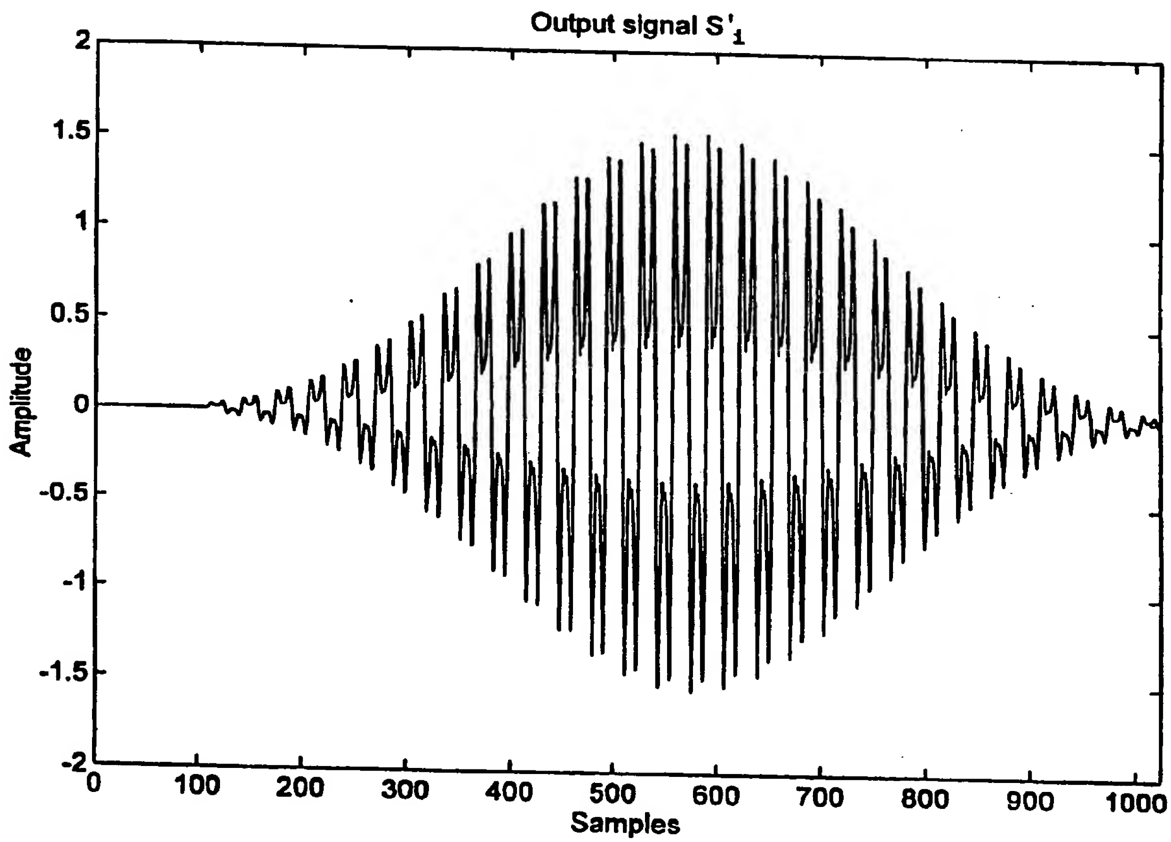

Output signal S'1
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
Amplitude
0
100
200
300
400
500
600
700
800
900
1000
Samples

FIG. 13

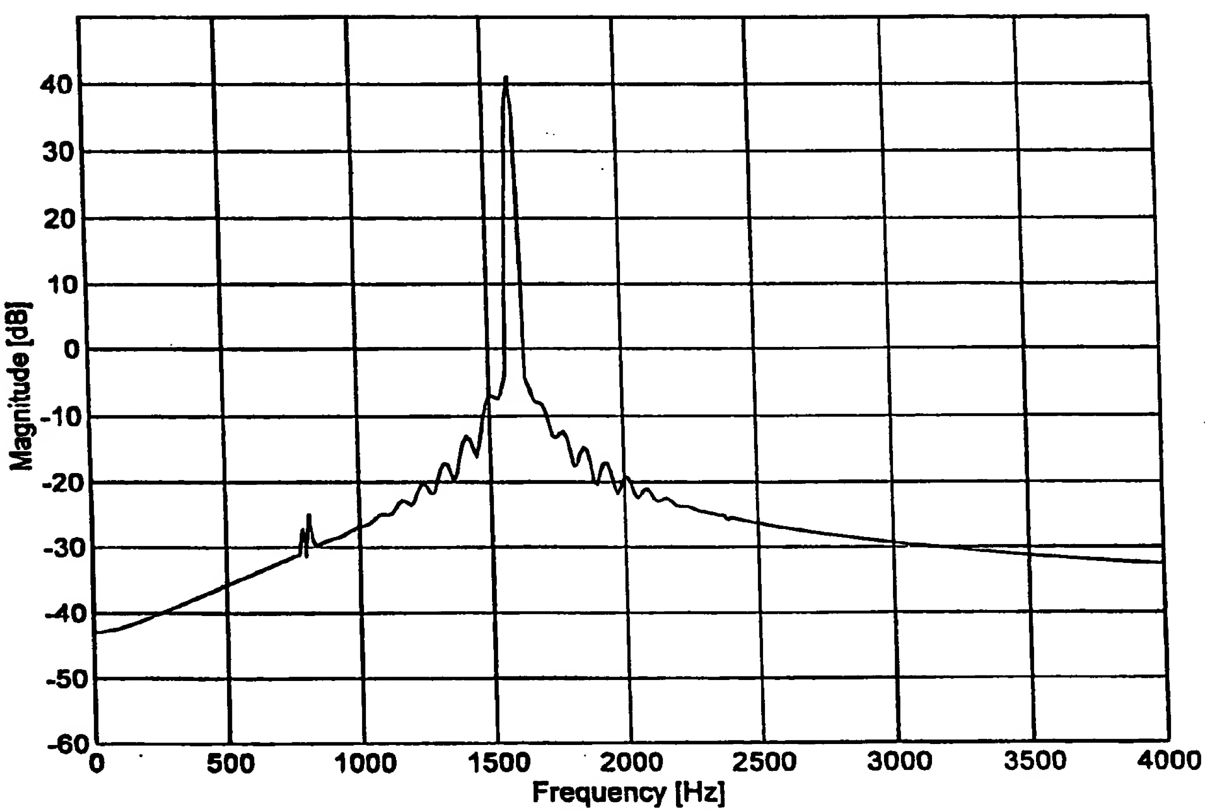
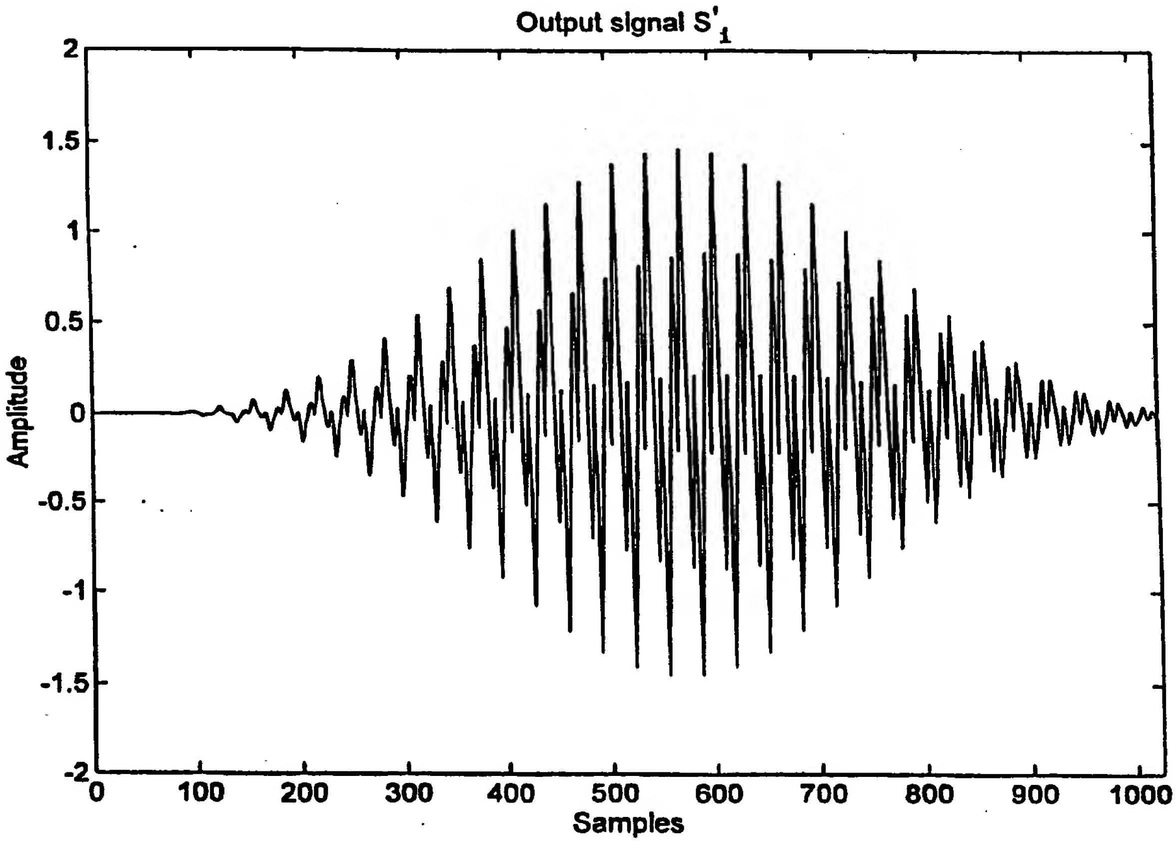

40
30
20
10
0
-10
-20
-30
-40
-50
-60
Magnitude [dB]
0
500
1000
1500
2000
2500
3000
3500
4000
Frequency [Hz]

FIG. 14



Output signal S'1
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
Amplitude
0
100
200
300
400
500
600
700
800
900
1000
Samples

FIG. 15

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